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PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant: Leon Examiner: To be assigned
Serial No.: 09/903901 Group Art Unit: 2661
Filed: 7/12/01 Docket No.: 975.349USW1
Title: RESPONSE TIME MEASUREMENT FOR ADAPTIVE PLAYOUT
ALGORITHMS

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- ☒ Certified copy of EP application, Serial Number PCT/EP99/00179, filed 14 January 1999;
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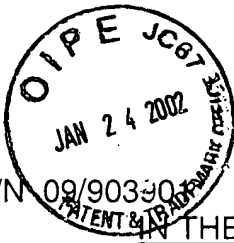
Respectfully submitted,

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6500 City West Parkway – Suite 100
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Date: December 17, 2001

By:

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S/N 09/903901

PATENT

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Applicant:	Leon	Examiner:	To be assigned
Serial No.:	09/903901	Group Art Unit:	2661
Filed:	7/12/01	Docket No.:	975.349USW1
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Dear Sir:

Enclosed is a certified copy of EP application, Serial Number PCT/EP99/00179,
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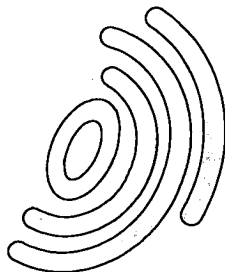
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Attestation

Die angehefteten Unterlagen stimmen mit der ursprünglich eingereichten Fassung der auf dem nächsten Blatt bezeichneten internationalen Patentanmeldung überein.

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28 NOV 2001

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Im Auftrag
For the President of the European Patent Office
Le Président de l'Office européen des brevets
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Patentanmeldung Nr.
Patent application no. PCT/EP 99/00179
Demande de brevet n°



Anmeldung Nr.: PCT/EP 99/00179
Application no.:
Demande n°:

Anmelder: 1. NOKIA TELECOMMUNICATIONS OY - Nokia Group, Finland
Applicant(s): 2. LÉON David - Tampere, Finland
Demandeur(s):

Bezeichnung der Erfindung:
Title of the invention: RESPONSE TIME MEASUREMENT FOR ADAPTIVE PLAYOUT
Titre de l'invention: ALGORITHMS

Anmeldetag:
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Staat:	Tag:	Aktenzeichen:
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Benennung von Vertragsstaaten : Siehe Formblatt PCT/RO/101 (beigefügt)
Designation of contracting states : See Form PCT/RO/101 (enclosed)
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RESPONSE TIME MEASUREMENT FOR ADAPTIVE PLAYOUT ALGORITHMS

5 FIELD OF THE INVENTION

The present invention relates to packet-switched networks used for real-time multimedia communications, and in particular to a system and a method for measuring the
10 response time for adaptive playout algorithms.

BACKGROUND OF THE INVENTION

Packet-switched networks are increasingly used for real-time
15 multimedia communications. Thus, there is a requirement that endpoints be able to recover from network impairments.

One of these impairments is called "jitter". Jitter can be considered in a wide sense. Herein, jitter is the variation
20 in the duration between the time a frame is captured by a transmitter audio card and the time it is received by a receiver. Therefore, it includes not only network jitter, i.e. variations in transmission delays, but also variations in processing delays.

25

Jitter is a severe audio stream impairment. In order to be understandable, audio streams must not be interrupted, or at least be interrupted as less as possible. If frames were played out as they arrive at the receiver, due to the jitter,
30 the playing would be constantly interrupted. Hence, arriving frames are not played out immediately, but kept in a so-called jitter buffer. A playout algorithm must then be

implemented in the receiver in order to determine the playout time of the received frames.

In its simplest form, the algorithm buffers the first
5 received frame for a predetermined time before playing it. Therefore, instead of interrupting the audio stream, an initial delay is applied to the stream.

The problem with such a method, however, is to decide how
10 long this buffer delay should be. A large delay will minimize the probability of an interruption but will cause a lack of interactivity between the end-users. Moreover, the packet delay distribution may be quite complex and variable over time. Thus, applying a fixed delay is satisfactory only in a
15 limited number of cases, e.g. in communications over a Local Area Network (LAN) with limited delay, but does not scale to more complex networks, particularly the Internet.

In order to overcome the above-mentioned problem, adaptive
20 algorithms have been introduced. Jitter adaptation is based on silence compression/expansion, wherein silence is a conversational device. In a conversation, silence indicates a speaker's expectation that his interlocutor starts talking. Therefore, silence can be expanded or compressed without
25 impairing the understandability. An adaptation algorithm estimates the jitter from packet arrival times and then modifies silence period lengths according to the latest estimate. For example, jitter adaptation algorithms based on this idea can be found in Sue B. Moon, Jim Kurose, Don
30 Towsley, "Packet audio playout delay adjustment: performance bounds and algorithms", Multimedia Systems, Springer Verlag 1998, pp. 17-28, and in Ramachandran Ramjee, Jim Kurose, Don

Towsley, Henning Schulzrinne, "Adaptive Playout Mechanism for Packetized Audio Applications in Wide-Area Networks", in Proceedings of the conference on computer communications, (IEE Infocom, Toronto, Canada), pp. 680-688, IEE Computer
5 Society Press, Los Alamitos, California, June 1994.

In the above-mentioned adaptive or adaptation algorithms, the received frames playout times are computed so as to achieve a good trade-off between buffering delay and residual drop
10 rate, which will be described later.

However, this adaptation scheme is not sufficient because it trades-off a drop percentage against an added buffering delay. What should be traded-off is the drop against the
15 response time which is defined as the time elapsed between the capture of a given frame of speech at one endpoint and its playout at an other endpoint plus the same quantity in the other direction. In accordance with the conventional adaptation scheme mentioned above, the added delay reflects
20 only partially the response time.

It is therefore an object of the present invention to overcome the aforementioned adaptation algorithm limitations and to allow a terminal to trade-off the response time
25 against the drop instead of the added buffering delay against the drop.

SUMMARY OF THE INVENTION

30 According to a first aspect of the present invention, this object is achieved by a system which comprises two endpoints communicating with each other by means of a packet-switched

network. The endpoints use adaptation algorithms for estimating jitter from packet arrival times and for modifying silence period lengths according to the latest estimate.

According to the present invention, the endpoints are able to

5 measure a response time ρ at a certain point of time and use it as a parameter in the adaptation algorithms.

According to a second aspect of the present invention, this object is achieved by a method for measuring a response time

10 ρ between two endpoints in a packet-switched network system, which comprises the steps of sending a response time request packet from a first endpoint to a second endpoint at a time s_r , receiving the response time request packet at the second endpoint at a time r_r , sending a response time indication
15 packet from the second endpoint to the first endpoint at a time s_i , receiving the response time indication packet at the first endpoint at a time r_i , and computing the response time ρ on the basis of the sending and receiving times in the first endpoint.

20

According to the present invention, a significant improvement in complex networks, in particular in Internet telephony quality, can be achieved.

25 Further developments of the present invention are defined in the respective appended subclaims.

In the following, a preferred embodiment of the present invention is described by taking into account the

30 accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 shows two endpoints communicating with each other, and

5 Fig. 2 shows a flowchart of the procedure for measuring the response time according to the preferred embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

10

The present invention is to be used in conjunction with adaptive or adaptation algorithms. It can be applied to any jitter adaptation algorithm based on silence compression/extension. At first, in order that the present
15 invention be more readily understood, a short description of these adaptation algorithms is given below.

In formalizing the algorithms, c is the capture time of a given frame for a transmitter and p is the playout time
20 scheduled by a receiver for this frame. If T is the transmission time of the considered frame, the following property $P1$ is obtained:

- if $p > c+T$, the frame is received before scheduled for
25 playout and can thus effectively be played out; and

- if $p < c+T$, the frame is dropped because it is not available in time.

30 When silence compression is used, a speech bit-stream is composed of active speech frames followed by silence frames. The received streams are conceptually fragmented in bursts

wherein a burst starts at the first frame of an active speech period and ends at the last frame of the following silence period. Thus, the burst contains not only the talkspurt (active speech), but also the silence period until the next
5 talkspurt.

All the (speech) frames belonging to the same burst are scheduled at an audio frame interval in order to avoid interruption during active speech, because such an
10 interruption is very annoying for users. Thus, the playout time $p_{i,j}$ for the i^{th} frame of the j^{th} burst can be written as:

$$p_{i,j} = p_{1,j} + (i-1)\delta,$$

15 wherein δ is the audio frame interval.

For the very first received frame (i.e. of the first burst) an initial $p_{1,1}$ is chosen (algorithm dependent).

20 When the next burst is received ($j+1$), the receiver may choose:

- either to keep the synchronization with the previous burst and in that case: $p_{1,j+1} = p_{I,j} + \delta$, with I being the last frame
25 of burst j ,

- or to adapt and use a new value $p_{1,j+1}$.

In the following it will be described how silence frame
30 suppression and addition are used to adjust to the playout discontinuities resulting from adaptation.

If the first frame of the burst $j+1$ is scheduled after the last frame of burst j , i.e. $p_{1,j+1} > p_{1,j}$, then the receiver plays out silence frames between the two playout times. It is
5 to be noted that, since silence may be added only as a multiple number of frames, $p_{1,j+1}$ cannot be set to the value as computed but only to a closest possible value.

Similarly, if the first frame of the burst $j+1$ is scheduled
10 before the last frame of burst j , i.e. $p_{1,j+1} < p_{1,j}$, this should reflect that the playout times have been previously overestimated. In that case, there should be some silence frames available in the playout buffer waiting for being played out. Some of those frames are discarded so that the
15 playout be as close as possible to the computed value.

From the above-given analysis, the adaptation algorithms exhibit the following property P2:

20 For certain adaptation points (usually the talkspurt start), the playout can be expressed as $p = r+B$, with r being a frame reception time and B a buffer delay chosen by the respective algorithm. For other packets (frames), the playout is synchronized with the previous packet playout, i.e. it is
25 obtained by adding an integral number of audio frame durations (audio frame intervals).

It is to be noted that, referring to property P1, the higher the value of B , the less the drop rate. The algorithms differ
30 only in the choice of B and the decision of when to adapt.

The present invention can be applied to any algorithm verifying the property P2.

The above-described jitter adaptation algorithms compute the
5 received frames playout times in order to achieve a good trade-off between buffering delay and residual drop rate.

However, as already mentioned, the adaptation scheme used by the jitter adaptation algorithms is not sufficient, because
10 it trades-off the drop percentage against the added buffering delay, as described in the foregoing. What should be traded-off is the drop against the response time. The added delay reflects only partially the response time. According to the present invention, a receiver is allowed to know the response
15 time at a certain point of time and to use it as a parameter in its adaptation algorithm, which will be described in the following.

In Fig. 1, two endpoints 1 and 2, i.e. two end-terminals, are
20 shown communicating with each other. Devices between the two end-users at the two endpoints 1 and 2, respectively, i.e. the endpoints 1 and 2 and a network (not shown), form a system according to the preferred embodiment of the present invention. The response time of the system at a given time
25 instant is defined as the time elapsed between the capture of a given frame of speech at one endpoint and its playout at the other endpoint plus the same quantity in the other direction.

30 As an illustration, it is supposed that one person (one end-user) asks another a question such as "how much is 2+2?". If the two persons were talking face to face it would take a

time T for the person to think of the result. If the two persons are now communicating through the system, it will now take a time $T+p$ to get the answer, with p being the response time as defined above.

5

To be precise, the value that matters to the end-users is therefore the response time as defined above and not the added buffer delay. As a consequence, it is this value which has to be traded off against the drop rate.

10

It can be demonstrated that, if two endpoints use a playout algorithm which exhibits the property P2, the following property P3 also holds:

15 - As long as no adaptation is done on either side (i.e. packet playout synchronized with that of the previous packet), the value of the response time remains constant.

- Whenever one of the endpoints performs adaptation, this
20 terminal can compute the increase or decrease of the response time due to the adaptation.

With respect to Fig. 1, c is the capture time and p is the scheduled playout time of a frame sent from the endpoint 1 to
25 the endpoint 2 in what is arbitrarily called the forward direction. Similarly, c' and p' are the same quantities in the reverse direction, i.e. c' is the capture time and p' is the scheduled playout time of a frame sent from the endpoint 2 to the endpoint 1 in the backward direction.

30

The response time as defined above is given as

$$\rho = (p-c)+(p'-c')$$

In order to demonstrate the property P3 the response time is calculated for two consecutive pairs of packets (frames) n
5 and n+1:

$$\rho_n = (p_n - c_n) + (p'_n - c'_n)$$

$$\rho_{n+1} = (p_{n+1} - c_{n+1}) + (p'_{n+1} - c'_{n+1})$$

10

If no adaptation is performed:

$$p_{n+1} = p_n + \delta \text{ and } c_{n+1} = c_n + \delta. \text{ Thus}$$

$$15 \quad p_{n+1} - c_{n+1} = p_n - c_n, \text{ and similarly}$$

$$p'_{n+1} - c'_{n+1} = p'_n - c'_n.$$

Therefore, if no adaptation is performed:

20

$$\rho_{n+1} = \rho_n.$$

It is now supposed that one of the endpoints chooses to adapt, for instance, the receiver on the forward path. In
25 that case, $p_{n+1} \neq p_n + \delta$.

The resulting variation in response time is then:

$$\Delta \rho_{n+1} = \rho_{n+1} - \rho_n = p_{n+1} - p_n - \delta.$$

30

This quantity can be calculated by the endpoint performing the adaptation.

The important consequences of property P3 are:

5

- As long as no adaptation is performed, changes in the network conditions do not produce any change on the response time value. For instance, a sudden increase in transmission times does not incur any increase in the response time.

10 However, fewer frames may arrive before their scheduled playout time and thus the drop rate may be increased.

- Since endpoints know the response time variation caused by adaptation, if they could measure the response time before
15 making adaptation, they could trade-off the response time against the drop rate.

For example, at a certain point, the receiver adaptation algorithm estimates that delaying the playout delay by a
20 further 200 ms would considerably decrease the loss or drop rate. If it knew that the response time at that time instant is 50 ms, then it could derive that the resulting response time will be 250 ms if it performs adaptation. It may then consider this value small enough and actually perform the
25 adaptation. On the other hand, if the response time is 800 ms before adaptation, it may consider that the resulting 1000 ms response time is too large and thus not adapt or adapt with a lower delay.

30 The present invention provides a system and a method for measuring the response time when the end-terminals use adaptation algorithms verifying the property P2, and

therefore allows the terminal or endpoint to trade-off the response time against the drop instead of the added buffering delay against the drop.

- 5 In the following, the measurement procedure according to the preferred embodiment of the present invention is described with reference to Figs. 1 and 2.

10 On the basis of property P3, any pair of frames (one in each direction) can be used to calculate the response time at a certain time instant, since the last adaptations were made on each side. For the sake of simplicity, the frames are used for which the last adaptation was made in the forward and reverse directions.

15

The playout times p and p' at the endpoint 2 and the endpoint 1, respectively, for those frames are given as:

$$p = r + D_p \text{ and}$$

20

$$p' = r' + D'_p,$$

25

with r and r' being the frame reception times of the endpoints 2 and 1, respectively, and D_p and D'_p being the respective adaptation playout delays.

It is assumed that s and s' are the times the corresponding frames were sent in the forward and reverse (backward) directions, respectively:

30

$$s = c + D_e \text{ and}$$

$$s' = c' + D'_E,$$

with c and c' being the respective capture times and D_E and D'_E being the respective encoding delays (the encodings need
5 not to be the same).

The response time $\rho = (p-c) + (p'-c')$ can thus be expressed as:

$$\rho = (r-s) + (r'-s') + (D_E + D_P + D'_E + D'_P), \text{ or}$$

10

$$\rho = T + T' + (D_E + D_P + D'_E + D'_P),$$

with T and T' being the respective frame transmission delays.

15 It is supposed that the terminal which sends packets along the forward path (the endpoint 1) wants to determine the response time. To that end, it sends a response time request packet (as a UDP packet in case RTP is used) to a port at the other endpoint 2 (S1 in Fig. 2) which was negotiated prior to
20 the transmission of the associated stream. Information carried in the request packet will be described later on.

Upon receipt of the request packet (S2 in Fig. 2), the endpoint 2 transmits immediately a response time indication
25 packet to a port at the endpoint 1 (S4 in Fig. 2) which was also negotiated in advance. Information carried in the indication packet will be described later on.

The request is sent at a time s_r from the endpoint 1 and is
30 received at a time r_r by the endpoint 2. The indication is sent at a time $s_i = r_r$ (or at least very close to) from the

endpoint 2 and is received at a time r_i by the endpoint 1 (S5 in Fig. 2).

The associated transmission times are:

5

$$T_r = r_r - s_r, \text{ and}$$

$$T_i = r_i - s_i.$$

10 The round-trip delay which can be measured by the endpoint 1 making the request is given as:

$$T_r + T_i = r_i - s_r.$$

15 In expressing the sum of the frame transmission delays $T+T'$ by:

$$\begin{aligned} T+T' &= (T-T_r) + (T'-T_i) + (T_r+T_i) \\ &= (r-r_r) + (s-s_r) + (r'-r_i) + (s'-s_i) + (T_r+T_i), \end{aligned}$$

20

the response time is given as:

$$\rho = (r-r_r) + (s-s_r) - (r'-r_i) + (s'-s_i) + (T_r+T_i) + D_E + D_P + D'_E + D'_P.$$

25 The idea to compute the response time is to see that some of the terms can be calculated by the endpoint 1 making the request and the remaining terms can be calculated by the endpoint 2 answering the request. The latter can therefore send the sum of the terms it knows in the indication packet.

30

The endpoint 1 (terminal making the request) knows:

- D_E
- D'_P
- $T_r + T_i$
- $r' - r_i$, since they are both measurable with the same clock.

5

s and s_r are also measurable using the request sender clock (the clock of the endpoint 1), but s is the sending time for which the receiver (the endpoint 2) performed adaptation. The sender, i.e. the endpoint 1, does not know a priori for which frame the receiver performed the latest adaptation. However, if the receiver indicates in the response time indication packet some information identifying that frame (for example its RTP timestamp in case RTP is used), the sender can lookup the corresponding sending time and make the computation $s - s_r$.

15 This, however, does not mean that the sender must keep in memory all the sending times of the packets it sends, since packets are sent at regular intervals. In case RTP is used, the sender can infer the difference in frame sending times from the frame RTP timestamps.

20

The endpoint 2 (terminal answering the request) knows:

- D'_E
 - D_P
- 25 - $r - r_r$

In addition, if the endpoint 1 making the request sends in the request packet some information identifying the latest frame for which it performed adaptation, the endpoint 2 can also calculate $s' - s_i$ (S_3 in Fig. 2) and can indicate this information in its indication packet.

30

Therefore, in step S6, the response time can be computed in the endpoint 1.

It is to be noted that the response time value remains valid
5 as long as none of the endpoints performs adaptation. If the
requested endpoint chooses to adapt between the time it sends
the response time indication and the time the other endpoint
receives it, then the computed response time might be an
outdated value. However, if the two endpoints agree on a
10 maximum adaptation step per unit of time, nevertheless an
upper bound on the response time can be derived.

A further point to be mentioned is that response time request
or indication packets might get lost. However, if requests
15 are made often enough, the response time value will be
updated at the next opportunity.

In the following, an example of an application of the present
invention is described.

20

It is supposed that an audio codec is used for which it is
considered that 20% is the maximum acceptable drop rate. It
is also supposed that experiments have been made to assess
the trade-off between drop and response time. For example, it
25 has been determined that a one-second response time and a 5%
drop is better than a two-second response time which would
lower the drop to 2%.

The endpoints have agreed that they are not allowed to
30 increase the response time by more than 100 ms every 10
seconds, and send a response time request every 5 seconds.

To the first frame they receive, the endpoints apply an initial buffer delay of for example 50 ms, and for the following talkspurts the following holds:

5 - If the drop rate is more than 20%, the buffer delay is increased to get 20%, no matter what value the measured response time has.

- If the drop rate is less than 20%, the measured response
10 time is traded-off (using an upper bound on the last measurement) against the residual drop rate.

According to the present invention, endpoints using any adaptation algorithm satisfying the property P2 are able to
15 measure the response time. In particular, the two endpoints need not use the same algorithm.

It is noted that an implementation of the present invention requires the definition of a complete protocol which
20 specifies the format of the response time request and indication packet (particularly the time format). The present invention is in no way limited to a particular protocol or implementation.

25 Thus, the present invention produces a significant improvement, for example, in Internet telephony quality.

While the invention has been described with reference to a preferred embodiment and an application example, the
30 description is illustrative of the invention and is not to be construed as limiting the invention. Various modifications and applications may occur to those skilled in the art

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without departing from the true spirit and scope of the invention as defined by the appended claims.

CLAIMS:

1. A system comprising two endpoints (1, 2) communicating
5 with each other by means of a packet-switched network, said
endpoints (1, 2) using adaptation algorithms for estimating
jitter from packet arrival times and for modifying silence
period lengths according to the latest estimate, wherein said
endpoints (1, 2) are able to measure a response time (ρ) at a
10 certain point of time and use it as a parameter in the
adaptation algorithms.
2. The system according to claim 1, wherein the adaptation
algorithms verify that for certain adaptation points a
15 playout (p) of a packet can be expressed as $p=r+B$, where r is
a packet reception time and B is a buffer delay chosen by the
algorithms, and that for other packets the playout is
synchronized with the previous packet playout.
- 20 3. The system according to claim 1, wherein said endpoints
(1, 2) can use different ones of said adaptation algorithms.
4. A method for measuring a response time (ρ) between two
endpoints in a packet-switched network system, comprising the
25 steps of:
 - sending (S1) a response time request packet from a first
endpoint (1) to a second endpoint (2) at a time s_r ;
 - receiving (S2) the response time request packet at said
second endpoint (2) at a time r_r ;
 - 30 sending (S4) a response time indication packet from said
second endpoint (2) to said first endpoint (1) at a time s_i ;

receiving (S5) the response time indication packet at said first endpoint (1) at a time r_i ; and

computing (S6) the response time (ρ) on the basis of the sending and receiving times in said first endpoint (1).

5

5. The method according to claim 4, wherein the response time request packet sent from said first endpoint (1) includes information identifying one of the packets which has been sent at a time s' by said second endpoint (2) and received at
10 a time r' by said first endpoint (1) since its latest adaptation, and wherein the response time indication packet sent from said second endpoint (2) includes information identifying one of the packets which has been sent at a time s by said first endpoint (1) and received at a time r by said
15 second endpoint (2) since its latest adaptation.

6. The method according to claim 5, wherein $s'-s_i$ is computed in said second endpoint (2) (S3) and the result is indicated in the response time indication packet.

20

7. The method according to claim 6, wherein in said calculating step the response time (ρ) is calculated according to the following expression:

25
$$\rho = (r-r_r)+(s-s_r)-(r'-r_i)+(s'-s_i)+(T_r+T_i)+D_E+D_P+D'_E+D'_P$$

wherein

D_E and D'_E are encoding delays of the first and second endpoints, respectively,

30 D'_P and D_P are adaptation playout delays of the first and second endpoints, respectively, and

$T_r = r_r - s_r$ and $T_i = r_i - s_i$, and

wherein

the quantities of D_e , D'_p , $T_r + T_i$, $r' - r_i$ and $s - s_r$ are known or can be computed in said first endpoint (1) and the

5 quantities of D'_e , D_p , $r - r_r$ and $s' - s_i$ are indicated in the response time indication packet.

ABSTRACT:

The present invention discloses a system which comprises two
5 endpoints (1, 2) communicating with each other by means of a
packet-switched network. The endpoints (1, 2) use adaptation
algorithms for estimating jitter from packet arrival times
and for modifying silence period lengths according to the
latest estimate. According to the present invention, the
10 endpoints (1, 2) are able to measure a response time (ρ) at a
certain point of time and use it as a parameter in the
adaptation algorithms.

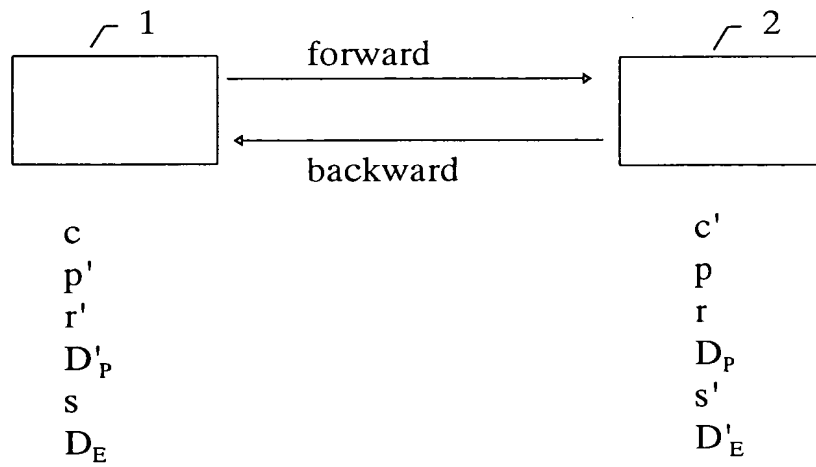


FIG. 1

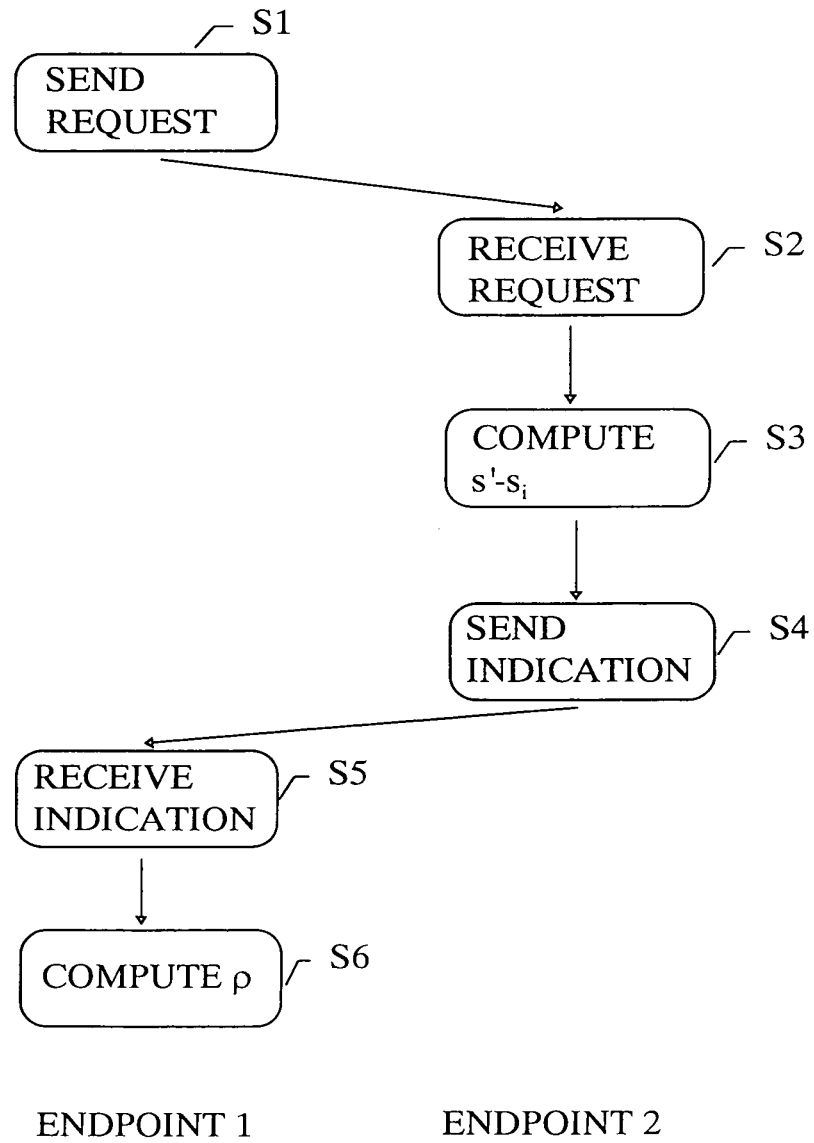


FIG. 2